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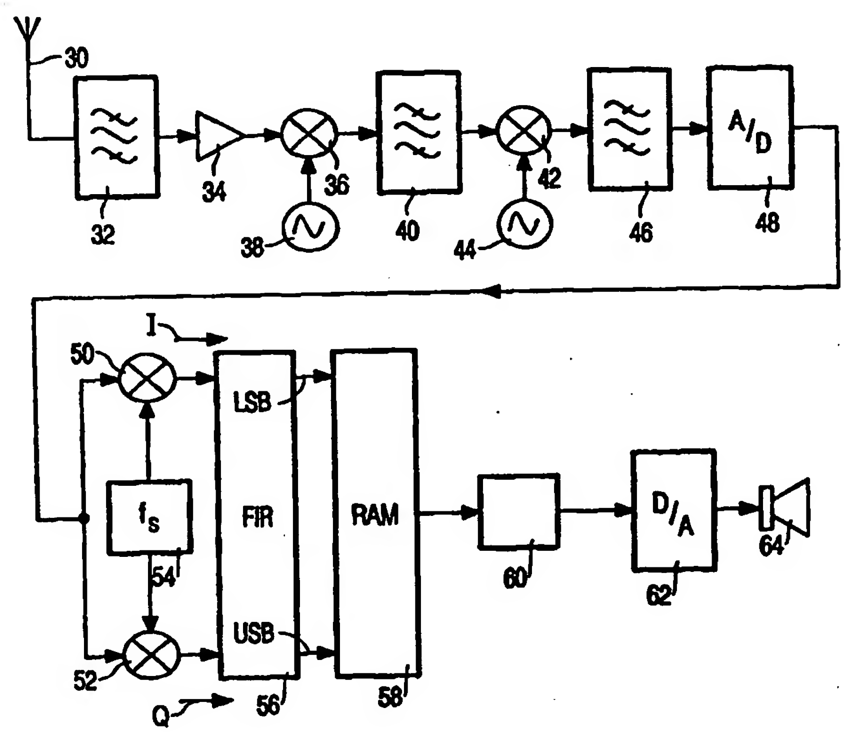
(54) Title: RECEIVER FOR INDEPENDENT SIDEBAND SIGNALS

(57) Abstract

A receiver for independent sideband (ISB) signals comprises means (30 to 54) for producing digitised quadrature related zero IF versions of the received signal which are applied to a complex FIR filter structure (56) comprising respective real low pass filters in which alternate coefficients (C_1 to C_{N-1}, C_0 to C_N) are non-zero. The FIR filter structure is illustrated in greater detail in Figure 3 (not shown). The respective upper and lower sidebands (USB, LSB) are recovered by obtaining the sum and difference of the outputs of the respective filters. The respective sideband signals (USB, LSB) are stored in RAM (58) and when it is desired to reproduce the stored signal it is expanded, equalised and converted to an analogue signal which is supplied to an audio transducer (64).



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DESCRIPTION

RECEIVER FOR INDEPENDENT SIDEBAND SIGNALS

Technical Field

The present invention relates to a receiver for receiving and demodulating independent sideband signals, which may comprise compressed analogue speech samples. Such a receiver may be used in a voice paging system.

Background Art

An independent sideband signal (ISB) consists of two independent single sideband signals. In its baseband representation the ISB signal is therefore a complex signal. Only a complex representation allows the negative frequencies to be different from positive. Often only one of the sidebands is interesting at a time. Traditionally, three different methods exist in the analogue domain to obtain either one of the sidebands as a real signal. The three methods are the filter method, the phasing method and the Weaver method. In the digital domain, the equivalent of the phasing method is usually applied. The broadband 90° shift needed can be achieved by a Hilbert transform.

A Hilbert transform shifts positive frequencies by -90° (introducing a lag) and negative frequencies by 90° (introducing a lead). The imaginary part of a complex signal lags the real part by 90°. If the imaginary part of a complex signal is Hilbert transformed, its positive frequencies show -180° phase shift whereas the negative frequencies show 0° phase shift. Adding the original real part of the signal to the Hilbert transformed part cancels out positive frequencies which are represented by the upper sideband (USB) in the baseband representation. Negative frequencies which are represented by the lower sideband (LSB) on the other side add constructively. Similarly, if the Hilbert transformed imaginary part is subtracted from the real part, the LSB cancels and the USB is obtained. In practice, the delay introduced in a

practical implementation of the Hilbert transform has to be compensated for in the real signal path. A symmetric N^{th} order FIR approach has a constant group delay of N/2, which can easily be introduced in the real signal path.

Figure 1 of the accompanying drawings shows such an approach based on a Hilbert transform FIR filter 10. In Figure 1, quadrature related real and imaginary baseband signals I and Q, respectively, are applied to respective in-phase and quadrature phase paths.

The Hilbert transform FIR filter 10 is provided in the quadrature phase path and comprises N series connected delay stages D1, D2, D3, D4 ... D(N-2), D(N-1), D(N), each introducing a delay of Z^{-1}. Multipliers M0, M1, M2, M3 ... M(N-2), M(N-1), M(N) are respectively coupled to the input of the first delay stage D1 and the outputs of all the delay stages D1 to D(N). Coefficients c_0, c_1, c_2, c_3 ... $c_{(N-2)}$, $c_{(N-1)}$, c_N are applied respectively to the multipliers M0 to M(N).

The products produced by the multipliers M0 to M(N) are summed in summing means 12 and appear on an output 14.

The in-phase path comprises a delay device 16 for delaying the real I signal by N/2.

The signal at the output 14 is summed with the signal from the delay device 16 in a signal combining stage 18 to provide the LSB signal which is filtered in band pass filter 22. The signal at the output 14 is subtracted from the signal from the delay device 16 in a signal differencing stage 20 to provide the USB signal which is filtered in a band pass filter 24.

As an alternative the Hilbert transform might also be done on the real part of the signal instead of the imaginary part. The output for LSB and USB are then exchanged. The order N of the Hilbert transform FIR filter depends on the unwanted sideband suppression needed. An ideal Hilbert transform is somewhat like an ideal brickwall filter which can only be accurately approximated by choosing N sufficiently large. The approximation suffers particularly around 0 Hz, where the phase response makes a jump from 90° to -90°. This means that the suppression of the unwanted sideband for low frequencies is not very high compared to the rest of the band.

The band pass filters 22, 24 are necessary to suppress out-of-band noise and any remaining carrier noise.

Disclosure of Invention

An object of the present invention is to reduce the complexity in recovering the sidebands of an ISB signal.

According to one aspect of the present invention there is provided a receiver for independent sideband (ISB) signals, comprising an input for a modulated ISB signal, means for providing quadrature related zero IF ISB signals and demodulating means for demodulating the zero IF ISB signals to produce respective upper and lower sideband signals, said demodulating means comprising first and second real digital filters with alternate non-zero coefficients, combining means for obtaining the sum of the outputs from the first and second filters as a lower sideband signal and differencing means for obtaining the difference between the outputs from the first and second filters as an upper sideband signal.

The present invention provides a receiver for independent sideband (ISB) signals, comprising an input for a modulated ISB signal, means for providing quadrature related zero IF ISB signals and demodulating means for demodulating the zero IF ISB signals to produce respective upper and lower sideband signals, said demodulating means comprising first and second real filters, inputs of the first and second real filters comprising respectively in-phase and quadrature phase components of the zero IF ISB signal, summing means having first and second inputs coupled to outputs of the first and second real filters and an output for providing one of the independent sideband signals and differencing means having first and second inputs coupled to the outputs of the first and second real filters, respectively, for subtracting the output of the second filter from the output of the first filter and an output for providing the other of the independent sidebands.

By providing real filters in both the quadrature related signal paths the complexity of the receiver is reduced by amongst other things not requiring

band pass filters to remove any out-of-band noise and any remaining carrier signal. This is made possible by the wanted frequency response being built into the ISB filtering arrangement.

The present invention is based on the realisation that by concentrating on the band of interest, effort need not be wasted in trying to produce an ideal phase jump around 0Hz, where the Hilbert transform exhibits difficulties.

In an embodiment of the present invention the first and second real filter filters comprise digital filters, each filter comprising N series connected delay stages, where N is an integer, in that outputs of the odd numbered stages of the first filter are coupled to respective multipliers to which respective coefficients are applied, outputs of the multipliers being coupled to summing means which provides a sum signal as its output and in that the input to the first stage and outputs of the even numbered stages of the second filter are coupled to respective multipliers to which respective coefficients are applied, outputs of the multipliers being coupled to summing means which provides a sum signal as its output. The coefficients applied to the multipliers of the first and second filters are real although they are derived from a complex filter.

According to a second aspect of the present invention there is provided an ISB filter comprising first and second real digital filters with alternate non-zero coefficients, combining means for obtaining the sum of the outputs from the first and second filters and differencing means for obtaining the difference between the outputs from the first and second filters.

Brief Description of Drawings

The present invention will now be described, by way of example, with reference to the accompanying drawings, wherein:

Figure 1 is a block schematic diagram of a known method of demodulating an ISB signal,

Figure 2 is a block schematic diagram of an embodiment of a receiver made in accordance with the present invention,

Figure 3 is a block schematic diagram of a FIR filter arrangement for

demodulating the respective sidebands,

Figure 4 is a spectrum of an ISB voice signal, and

Figures 5A, 5B and 5C illustrate respectively characteristics of an example of a low pass filter, and of a USB filter and a LSB filter derived from low pass filter coefficients.

In the drawings the same reference numerals have been used to indicate corresponding features.

Mode for Carrying Out the Invention

Referring to Figure 2, the receiver comprises an antenna 30 which is coupled by a band pass filter 32 and an rf amplifier 34 to one input of a mixer 36. A first local oscillator 38 is coupled to a second input of the mixer 36 and is used to frequency down convert the input signal to a first IF signal. The first IF signal is filtered in a band pass filter 40 and is frequency down converted to a lower, second IF signal in a mixer 42 using a signal derived from a second local oscillator 44. The second IF signal which has a frequency around 10kHz is applied to an anti-alias filter 46 which reduces the bandwidth of the signal and removes all dc components and the output is digitised in an analogue to digital converter 48 operating at 4 times the second IF. The digitised signal is frequency down converted to quadrature related real and imaginary base band signals I and Q using multipliers 50, 52 and a frequency source 54 which produces 90 degree phase shifted outputs. The I and Q outputs are applied to a FIR filter arrangement 56 to be described in greater detail later with reference to Figure 3. Outputs from the arrangement 56 comprise the respective upper and lower sideband signals USB and LSB. These are stored in a RAM 58 in readiness to be read-out by the user. In reading out USB and LSB signals they are applied to an expand/equalise stage 60 which reassembles the voice message, adjusts pitch and corrects amplitude fluctuations. The reassembled message is converted into an analogue signal in a digital to analogue converter 62 and the output is applied to an audio transducer 64.

The FIR filter arrangement 56 shown in Figure 3 comprises 2 real low pass filters processing the I and Q signals, respectively.

In the case of the I signal path, the filter comprises series connected delay stages DI1, DI2 ... DI(N). Multipliers M1, M3, ... M(N-3), M(N-1) are coupled to the outputs of the odd-numbered delay stages DI1, DI3, ... DI(N-3), DI(N-1). Real coefficients c_1, c_3, ... c_{N-3} and c_{N-1} are applied to the multipliers M1, M3 ... M(N-3), M(N-1), respectively. The products produced are combined in a summing stage 12A which has an output 14A.

For the Q signal path, the filter has a degree of similarity in that it comprises series connected delay stages DQ1, DQ2 ... DQ(N-2), DQ(N-1), DQ(N), multipliers M0, M2, M4 ... M(N-2), M(N) and a summing stage 12B for combining the outputs from the multipliers and providing an output signal on output 14B. However the differences are that the multipliers M0 ... M(N) are connected to the input of the first delay stage DQ1 and to the outputs of even - numbered delay stages DQ2, DQ4 ... DQ (N-2), DQ(N). Also the real coefficients c_0, c_2, c_4 ... c_{N-2}, c_N are different from those applied to the multipliers of the other filter.

Compared to Figure 1, Figure 3 has 2 real low pass filters each having a series of delay stages but the number of multipliers per filter is of the order half that used in the Hilbert filter shown in Figure 1. By having 2 real low pass filters, the total number of multipliers is no more than is used in the Hilbert filter and the outputs from the summation stages 14A, 14B can be used without any additional bandpass filtering which is necessary using the arrangement shown in Figure 1.

The ISB signals are often audio signals, the frequency range of which start at some 100 or 200 Hz and reach up to the kHz range incorporating a total bandwidth of f_{bw}. If an N[th] order FIR filter with the coefficients $\{c_0, c_1, \ldots, c_N\}$ has a lowpass frequency response with cutoff frequency of $f_{bw}/2$, then the filters deriving LSB and USB in the interesting bandwidth from $f_s/4 - f_{bw}/2$ to $f_s/4 + f_{bw}/2$, where f_s is the sampling frequency of the filters, can easily be constructed with the same coefficients as shown in Figure 3. The

filter order of the filters shown in Figure 3 is therefore smaller than the filter order of the Hilbert transform for the same unwanted sideband attenuation. Although two filters for I and Q are used, still only N+1 multiplications have to be performed. Having a lower order N means less additions and multiplications are necessary and the group delay of the circuit is smaller. Moreover, additional filtering for noise shaping for example can be avoided using this approach.

A real lowpass filter can be used to derive a complex bandpass filter by mixing the coefficients of the FIR filter with an exponential as in equation (1) below.

$$\tilde{c}_k = c_k \cdot e^{j(2\pi . f \frac{k}{f_s} + p)} \qquad (1)$$

where $\tilde{c}_k$ is the coefficient of the complex bandpass FIR filter, c_k is the k_{th} coefficient of the original lowpass FIR filter, f is the mixing or shifting frequency, and f_s is the sampling frequency and ρ is an arbitrary phase. By choosing ρ a multiple of

$$\frac{2\pi . f}{f_s} \qquad (2)$$

the amplitude response of the filter stays invariant. Equation (1) can therefore be written as

$$\tilde{c}_k = c_k \cdot e^{j(2\pi . f \frac{k+1}{f_s})} \qquad (3)$$

The special case of

$$f = \frac{f_s}{4} \qquad (4)$$

leads to the coefficients for the USB filter

$$\tilde{c}_k = c_k \cdot e^{j2\pi \frac{k+1}{4}} = c_k \cdot j^{k+1} \tag{5}$$

Similarly, shifting the filter response in the negative direction

$$(f = -\frac{f_s}{4}) \tag{6}$$

we get the coefficients of the LSB filter as

$$\tilde{c}_k = c_k \cdot j^{-(k+1)} \tag{7}$$

The two coefficient sets for USB and LSB are therefore

$$(jc_0,\ -c_1,\ -jc_2,\ c_3,\ jc_4,\ -c_5,\ -jc_6,\ c_7 \ldots,\ j^{N+1}c_N) \tag{8}$$

and

$$(-jc_0,\ -c_1,\ jc_2,\ c_3,\ -jc_4,\ -c_5,\ jc_6,\ c_7 \ldots, -j^{N+1}c_N) \tag{9}$$

respectively. Two special properties can be observed. Firstly, each coefficient is either real or imaginary but not mixed. Secondly, the coefficient sets for the USB and the LSB filter differ only in every second place and only in their signs. Only the imaginary coefficients have a reverse sign. Exploiting these properties leads to a structure as shown in Figure 3.

By way of example, in a special case of making N = 8, it was found that 40 dB of unwanted sideband suppression can be obtained by applying a 16th order FIR filter as the low pass filter. The filter had been designed using an equiripple approach.

For the sake of illustration Figure 4 illustrates an example of an ISB signal and by inspection it will be noted that the USB and LSB are different.

Figure 5A illustrates the frequency response of a prototype low pass filter for the case of N=16, f_s = 6.4kHz and f = 1.6kHz.

Figures 5B and 5C illustrate, respectively, the frequency response of the USB filter and LSB filter derived from the low pass filter coefficients.

ISB filters may be used in applications whenever AM - SSB signals are present. The ISB filter may be implemented either in hardware from commercial parts, programmed in an FPGA or as a full - custom ASIC or software running on a DSP.

From reading the present disclosure, other modifications will be apparent to persons skilled in the art. Such modifications may involve other features which are already known in the design, manufacture and use of ISB receivers and filters and component parts thereof and which may be used instead of or in addition to features already described herein.

Industrial Applicability

Independent sideband communications systems such as voice paging systems.

CLAIMS

1. A receiver for independent sideband (ISB) signals, comprising an input for a modulated ISB signal, means for providing quadrature related zero IF ISB signals and demodulating means for demodulating the zero IF ISB signals to produce respective upper and lower sideband signals, said demodulating means comprising first and second real digital filters with alternate non-zero coefficients, combining means for obtaining the sum of the outputs from the first and second filters as a lower sideband signal and differencing means for obtaining the difference between the outputs from the first and second filters as an upper sideband signal.

2. A receiver for independent sideband (ISB) signals, comprising an input for a modulated ISB signal, means for providing quadrature related zero IF ISB signals and demodulating means for demodulating the zero IF ISB signals to produce respective upper and lower sideband signals, said demodulating means comprising first and second real filters, inputs of the first and second real filters comprising respectively in-phase and quadrature phase components of the zero IF ISB signal, summing means having first and second inputs coupled to outputs of the first and second real filters and an output for providing one of the independent sideband signals and differencing means having first and second inputs coupled to the outputs of the first and second real filters, respectively, for subtracting the output of the second filter from the output of the first filter and an output for providing the other of the independent sidebands.

3. A receiver as claimed in Claim 2, characterised in that the first and second real filters are digital filters in which alternate coefficients are non-zero.

4. A receiver as claimed in Claim 2, characterised in that the first and second real filter filters comprise digital filters, each filter comprising N

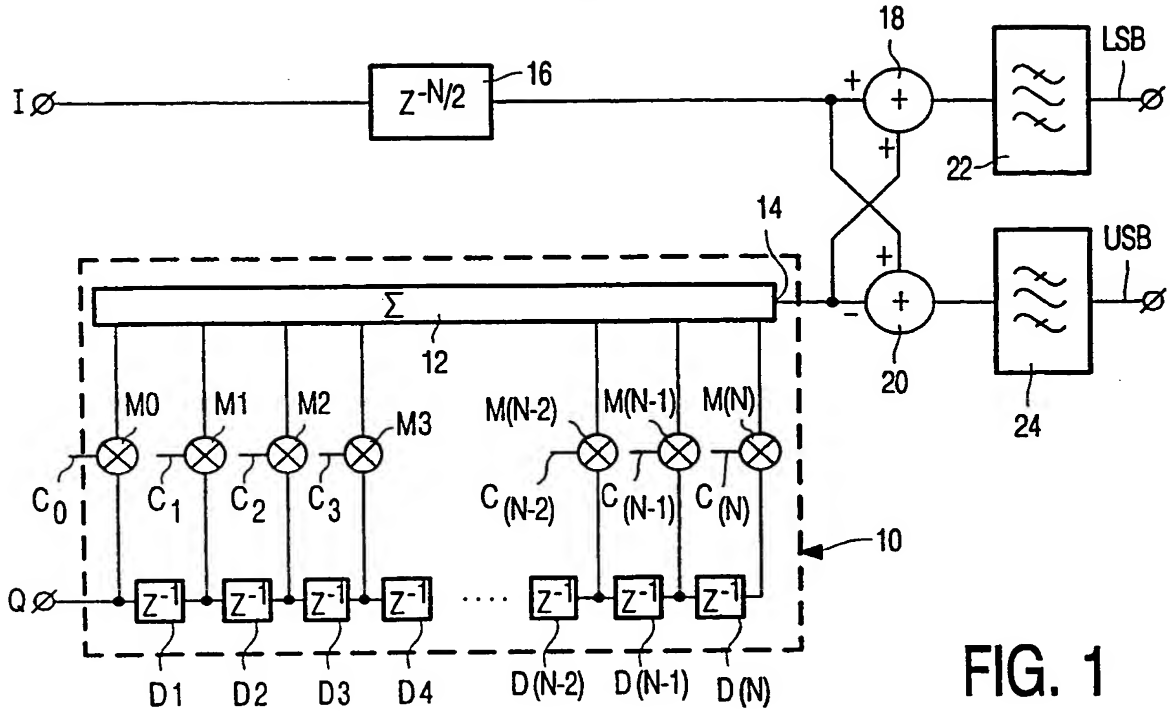
series connected delay stages, where N is an integer, in that outputs of the odd numbered stages of the first filter are coupled to respective multipliers to which respective coefficients are applied, outputs of the multipliers being coupled to summing means which provides a sum signal as its output and in that the input to the first stage and outputs of the even numbered stages of the second filter are coupled to respective multipliers to which respective coefficients are applied, outputs of the multipliers being coupled to summing means which provides a sum signal as its output.

5. A receiver as claimed in Claim 4, characterised in that the coefficients applied to the multipliers of the first and second filters are real.

6. An ISB filter comprising first and second real digital filters with alternate non-zero coefficients, combining means for obtaining the sum of the outputs from the first and second filters and differencing means for obtaining the difference between the outputs from the first and second filters.

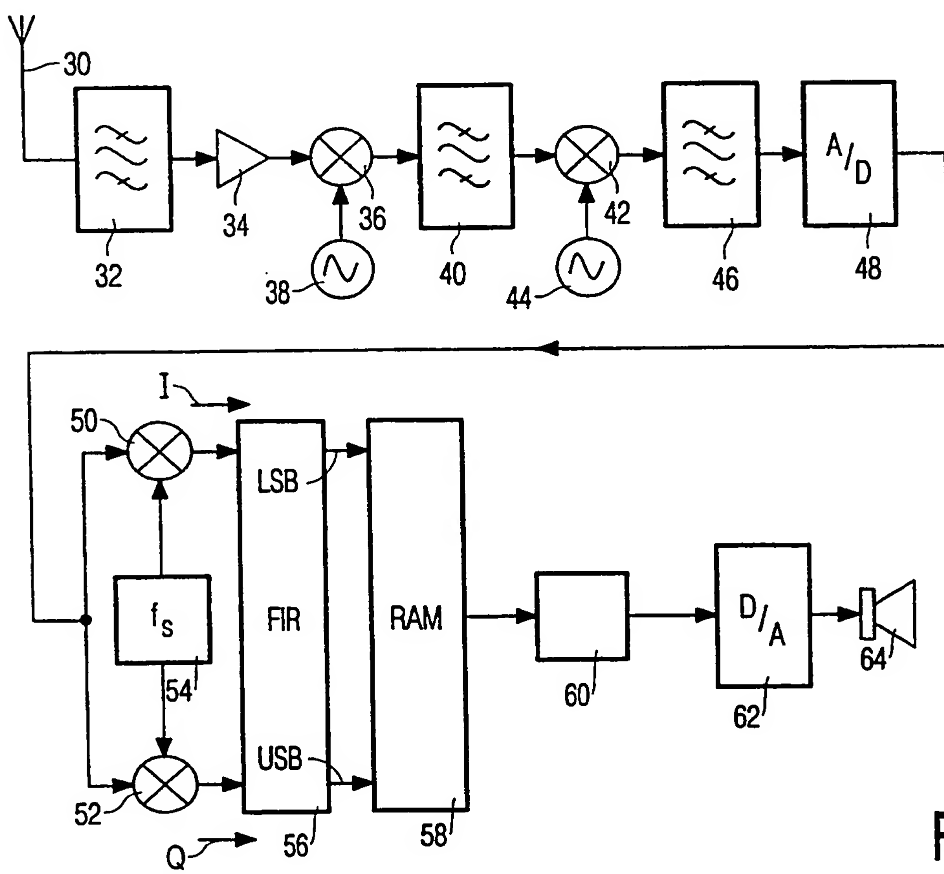
7. An ISB filter as claimed in Claim 6, characterised in that the first and second real filter filters comprise digital filters, each filter comprising N series connected delay stages, where N is an integer, in that outputs of the odd numbered stages of the first filter are coupled to respective multipliers to which respective coefficients are applied, outputs of the multipliers being coupled to summing means which provides a sum signal as its output and in that the input to the first stage and outputs of the even numbered stages of the second filter are coupled to respective multipliers to which respective coefficients are applied, outputs of the multipliers being coupled to summing means which provides a sum signal as its output.

8. An ISB filter as claimed in Claim 7, characterised in that the coefficients applied to the multipliers of the first and second filters are real.



I
$Z^{-N/2}$
16
18
+
+
+
LSB
22
14
USB
Σ
12
–
20
24
M0
M1
M2
M3
M(N-2)
M(N-1)
M(N)
C_0
C_1
C_2
C_3
$C_{(N-2)}$
$C_{(N-1)}$
$C_{(N)}$
10
Q
Z^{-1}
D1
D2
D3
D4
D(N-2)
D(N-1)
D(N)

FIG. 1



30
32
34
36
38
40
42
44
46
A/D
48
I
50
LSB
FIR
USB
RAM
f_S
54
52
Q
56
58
60
D/A
62
64

FIG. 2

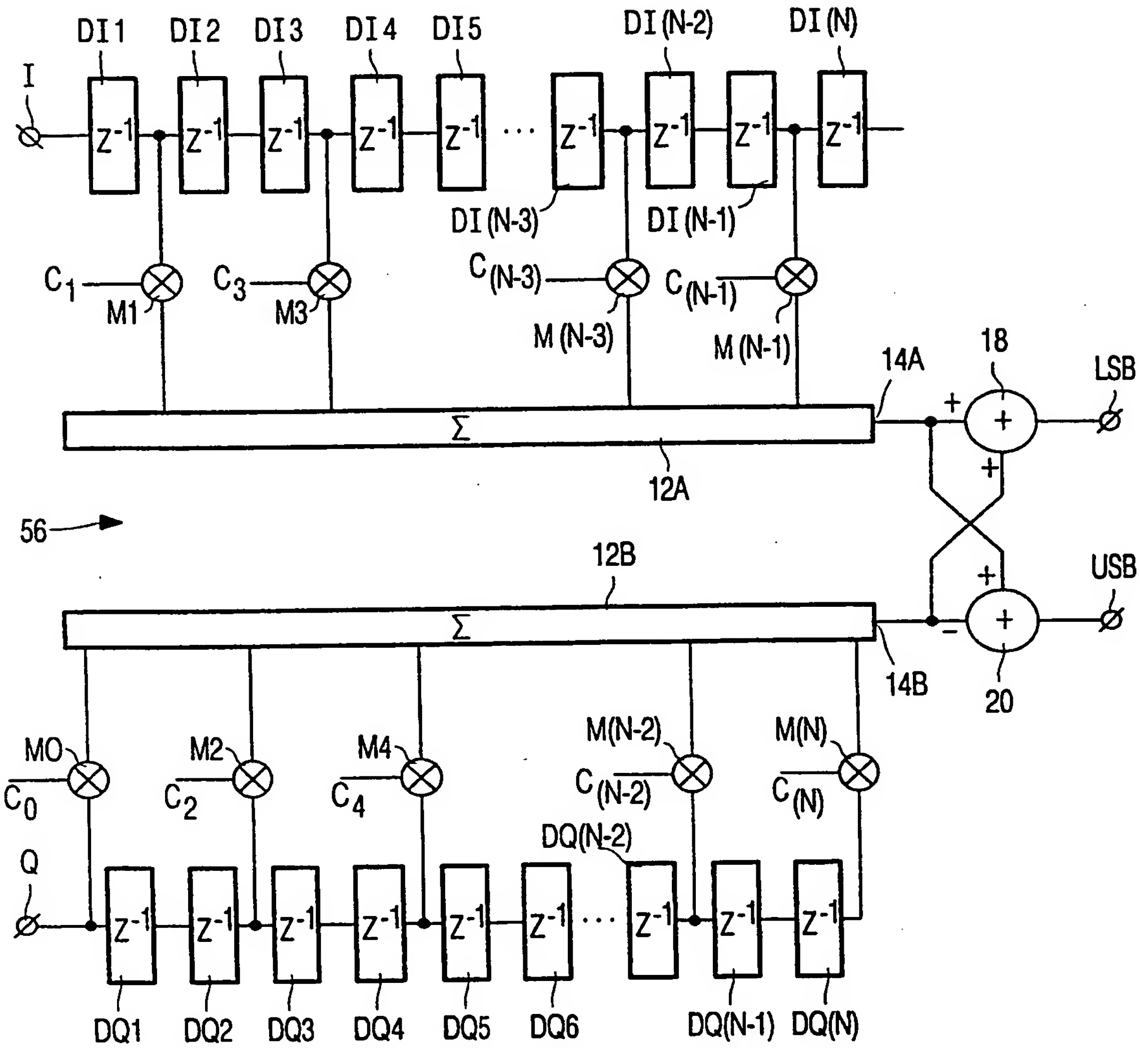

DI1
DI2
DI3
DI4
DI5
DI(N-2)
DI(N)
I
DI(N-3)
DI(N-1)
C1
M1
C3
M3
C(N-3)
M(N-3)
C(N-1)
M(N-1)
14A
18
LSB
Σ
12A
56
12B
USB
Σ
14B
20
MO
C0
M2
C2
M4
C4
M(N-2)
C(N-2)
M(N)
C(N)
DQ(N-2)
Q
DQ1
DQ2
DQ3
DQ4
DQ5
DQ6
DQ(N-1)
DQ(N)

FIG. 3

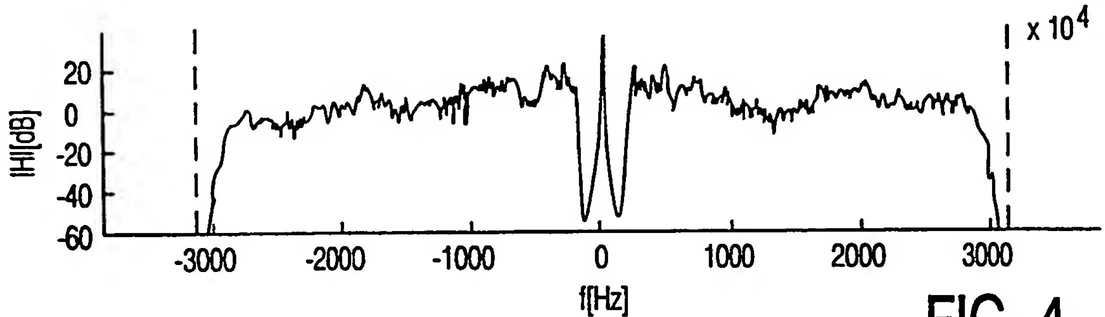


FIG. 4

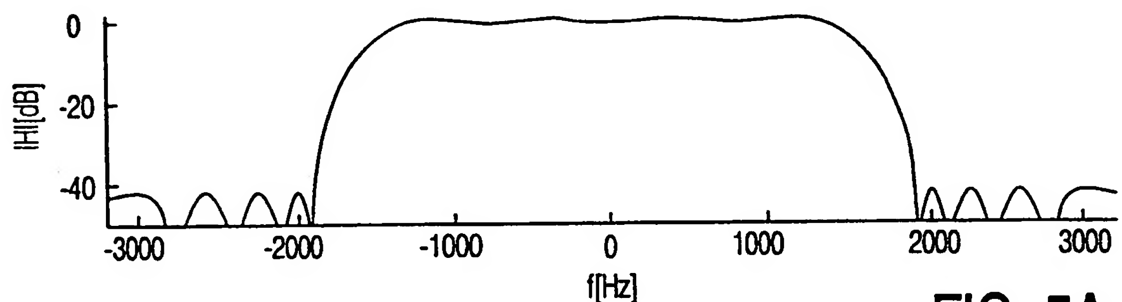


FIG. 5A

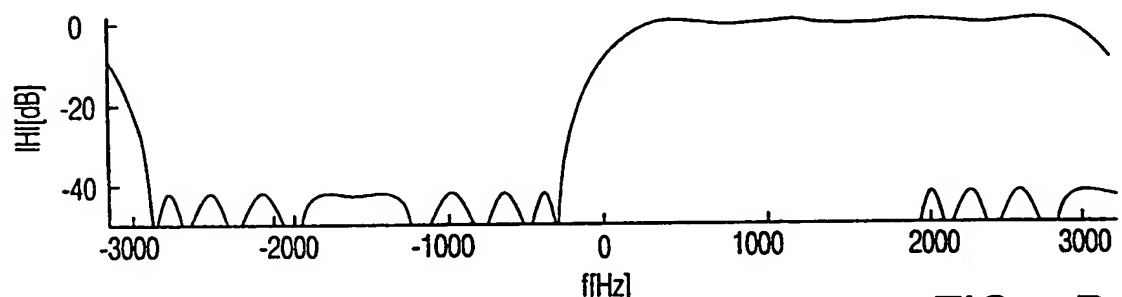


FIG. 5B

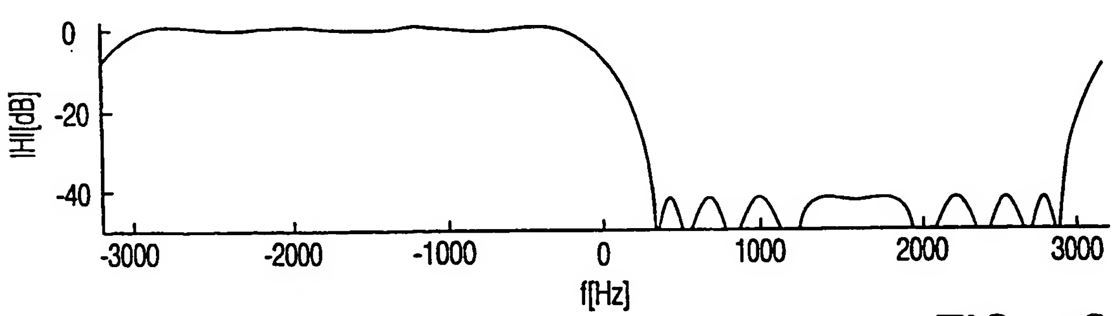


FIG. 5C

INTERNATIONAL SEARCH REPORT

International application No.
PCT/IB 98/00340

A. CLASSIFICATION OF SUBJECT MATTER

IPC6: H04L 27/22, H04B 14/04, H04N 5/44
According to International Patent Classification (IPC) or to both national classification and IPC

B. FIELDS SEARCHED

Minimum documentation searched (classification system followed by classification symbols)

IPC6: H04L, H04B, H04N

Documentation searched other than minimum documentation to the extent that such documents are included in the fields searched

SE,DK,FI,NO classes as above

Electronic data base consulted during the international search (name of data base and, where practicable, search terms used)

C. DOCUMENTS CONSIDERED TO BE RELEVANT

Category*	Citation of document, with indication, where appropriate, of the relevant passages	Relevant to claim No.
X	EP 0509725 A2 (BOSE CORPORATION), 21 October 1992 (21.10.92), figure 6, abstract --	1-8
A	US 5264937 A (LAUREN A. CHRISTOPHER), 23 November 1993 (23.11.93), column 1, line 19 - line 32, abstract --	1-8
A	US 5235647 A (LEON M. VAN DE KERKHOF), 10 August 1993 (10.08.93), abstract --	1-8
A	US 4361893 A (GEORGES BONNEROT), 30 November 1982 (30.11.82), figure 7, abstract -- --------	1-8

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Date of the actual completion of the international search	Date of mailing of the international search report
25 Sept 1998	28 -09- 1998
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INTERNATIONAL SEARCH REPORT

Information on patent family members

27/07/98

International application No.

PCT/IB 98/00340

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